Description

Handling of early media II

The invention relates to a method for selecting "early media"
user data transmitted from at least one called subscriber B to
a calling subscriber A via at least one telecommunication
network during initiation of a call.

The Session Initiation Protocol (SIP) is a signaling protocol
that can be used for call control, for example, of telephone
conversations. SIP is standardized by the IETF in RFC 3261 and
in an older version in RFC 2543. To describe the switched
communication connection, SIP uses the Session Description
Protocol (SDP), IETF RFC 2327, in a manner described in IETF
RFC 3264. SIP, together with the negotiated full user data
connections (e.g. speech connections), is usually transmitted
using the Internet protocol. SIP is used in the above-
described manner in, for example, the Internet Multimedia
Subsystem (IMS) of a mobile radiocommunication network
standardized by 3GPP or 3GPP2.

During initiation of a call from the SIP terminal of a caller
A to a called user B, the SIP signaling can be redirected by
switching nodes or "proxies". The proxies are permitted to
redirect an incoming message which presents a request by the
user A for a connection to B (an INVITE request) to a
plurality of other proxies or SIP terminals simultaneously or
sequentially - for example, in order to search for the user B.
Because the last-mentioned proxies can branch the message when
redirecting it, a tree-like branching of the message can
occur. This branched redirection of messages is referred to in
SIP as "forking".

When the INVITE message reaches a terminal of user B, this
terminal can respond with what is called a "1xx provisional
response" which can serve, for example, to negotiate the media
(e.g. speech, video) used for the communication connection and
their coding, or to indicate that the user B is being alerted
(for example, by the ringing of a SIP telephone). In the event
of forking, it can happen that a plurality of terminals send
such provisional responses - for example, if a plurality of
SIP telephones ring simultaneously. To conclude the initiation
of the communication relationship between a terminal of caller
A and a terminal of the called party B, the latter terminal
responds with what is called a "2xx final response", for
example, when user B has taken the SIP telephone off the hook.
A plurality of terminals of B can send such final responses,
for example, if a plurality of ringing SIP telephones are
taken off the hook. Accordingly, it can happen that the
terminal of A receives provisional responses and/or final
responses from a plurality of terminals of B. Each terminal of
B provides all the messages it sends as responses to A with
the same unique identification. If SIP response messages with
a new identification reach the terminal of A, the terminal of
A learns therefrom that it is communicating with a new
terminal point. In this case the SIP refers to a "dialogue"
existing between the terminal of A and the responding terminal
of B. Before A (and/or B, if applicable) has received a final
response for a dialogue, this is referred to as an "early
dialogue", the subsequent state as an "established dialogue".

It can happen that before the end of initiation of the
communication relationship, the terminals of A and B exchange
media (user data) which is referred to as "early media". For
example, as in a conventional telephone network, ring tones

and announcements may be transmitted, preferably in the direction from B to A. For a telephone network with SIP signaling, support by an "early media" transmission is especially important if the network is connected to a conventional telephone network.

If a plurality of dialogues are established in (/with) terminal A during initiation of the communication relationship from A to B as a result of forking, A may also receive media (user data), in particular early media, from different terminals B, B'. The terminal of A must represent the media in a suitable fashion. For example, it is possible that different incoming video streams are displayed in separate windows on a display screen. Frequently, however, it is appropriate to select only one incoming media stream and to reject the remaining media streams - for example, because the display screen in a mobile terminal is too small to show a plurality of windows, or because the superposing of different ring tones or announcements would make the content unintelligible.

Information on the corresponding SIP dialogues might be criteria permitting the selection of a suitable media stream (user data stream) for representation:
- If an early dialogue becomes an established dialogue through receipt of the first SIP final response, it is appropriate to select the corresponding media stream.
- It may be appropriate to select the early media that corresponds to the early dialogue last established. This is especially the case if the proxies initiate forking in a sequential manner. If the terminal sends a negative response, or if after a given time the communication relationship with it has not been established, for example, because no user has gone "off-hook", a proxy redirects the INVITE request to a

different terminal. The IETF specifies methods which will
enable terminal A to request a proxy to search only
sequentially (draft-ietf-sip-callerprefs).
- Terminal A can terminate dialogues using SIP signaling - for
example, because it is able to support only a limited number
of dialogues. The corresponding media can, however, continue
to be received for a certain time because of the delay times
of signaling and media. It is desirable to suppress the media
during this transition time.

The information contained in SIP and SDP does not always
unambiguously allow a SIP dialogue to be correlated with the
corresponding media stream. In particular, the terminal of
caller A selects an IP address and port, for example, a UDP
port (see IETF RFC 768) to receive the media streams before it
sends the INVITE request containing this information. All
incoming media are therefore received at the same IP address
and the same port. They can be distinguished by means of the
parameter "source IP address" in the IP header and "source
port" in the UDP header of the packets received, i.e. the IP
address and the port from which the packets were sent.
However, according to RFC 3264, no information on this source
IP address and source port is contained in SIP/SDP, but only
on the destination IP address and the destination port, i.e.
the IP address and port to which the packets were sent.

When SIP forking was designed, the interaction with early
media was at first not considered, since early media occur in
a SIP network only in special cases, for example, in
conjunction with a conventional telephone network.

The handling of early media (user data) in the case of forking
is currently being discussed in the IETF SIPPING Working

Group. The "draft-camarillo-sipping-early-media" draft
proposes that separate communication connections for early
media user data be negotiated using SIP, terminal B acting as
caller in the communication connections for early media when
it receives a call from A for the actual user connection and
initially enters into an early dialogue regarding this call
for the user connection with A. However, this has the
disadvantage that considerably more SIP messages must be
exchanged, leading to delay in initiating the call and higher
resource demand, especially when transmitting via an air
interface with narrow bandwidth. In addition, it might
possibly be necessary to reserve separate transmission
resources for early media and for the actual user connection.

In the "draft-ietf-mmusic-sdp-srcfilter" the IETF MMUSIC
Working Group proposes to introduce into SDP a parameter which
allows expression of the source IP address and the source UDP
port from which a recipient wishes to receive packets. This
information is useful in configuring interposed firewalls.
However, this parameter is unsuitable for correlating SIP
dialogues and media streams because it presupposes that the
recipient already knows the source IP address and the source
UDP port. Moreover, the use of this parameter in H.248
signaling has not so far been described.

It is the object of the present invention to make it possible
to select (early media) user data as efficiently as possible
during SIP call initiation signaling. This object is achieved
by the subject matter of the independent claims.
The transmission according to the invention of called
subscriber transmission address data (IP b, port b for
subscriber B or IP b', port b' for subscriber B'), in addition
to the called subscriber reception address data (IP B, port B

for subscriber B, etc.) known to be transmitted in any case, in a response message (provisional response and/or final response of a destination to a calling subscriber) enables the calling subscriber A to use this received destination transmission address data to select efficiently from early media user data of different called subscribers (B,B') received by it.

The fact that, according to the invention, called subscriber transmission address data (IP b, port b) is derived from a response message (provisional response 9, 10; 11, 12; final response 17) of a called subscriber (B; B') received by the calling subscriber (A) and containing called subscriber transmission address data (IP b, port b) in addition to called subscriber reception address data (IP B, port B) can mean, for example, that this called subscriber transmission address data (IP b, port b) can be noted by the calling subscriber (A) or (temporarily) stored for later selection.

Advantageous embodiments of the invention are apparent from the claims and from the following description of an embodiment. In the drawings:

Fig. 1   represents diagrammatically the signaling during
           initiation of a call and transmission of early media
           user data.

Cellular mobile radiocommunication networks (such as GSM, 3G CDMA2000, TDSCDMA, etc.) and fixed networks, and associated terminals and signaling procedures (SIP, SDP), are known *per se* to those skilled in the art (see, for example, specifications in www.3gpp.org).

Fig. 1 shows a calling subscriber A, including a SIP terminal
A connection part and a SIP terminal A signaling part, which
communicates via a mobile radiocommunication network
(represented here only by a SIP proxy necessary for an
understanding of the invention) with a called subscriber (=B)
including a SIP terminal B, and a called subscriber (=B')
including a SIP terminal B', using a SIP protocol to initiate
a telecommunication connection (e.g. speech connection, etc.).
The SIP terminal A connection part may be, for example, a so-
called "IM-MGW" and the SIP terminal A signaling part may be a
so-called "MGCF", the SIP proxy may be a so-called "S-CSCF"
and the SIP terminals B and B' may be so-called "UE"s. For
simplicity a number of SIP messages, for example, "100
Trying", PRACK and 200 OK (PRACK) have been omitted.

In the example illustrated, following a message 1 from the SIP
terminal A signaling part to the SIP terminal A connection
part, initiation of a telecommunication connection (for
example, for a speech connection or other user data
connection) is attempted, the messages 3-7, 9, 10, 13 being
exchanged between the calling subscriber A and the called
subscriber B (via a signaling network / via the SIP proxy)
before the user B goes off-hook (step 15) at the called
subscriber terminal B.

The SIP terminal A connection part selects the address to be
used by the SIP terminal A for future reception (IP address of
A (IP A) and port number of A (port A)), transfers these in
step 3 to the SIP A signaling part which, in step 4, sends a
SIP INVITE message specifying the terminal A reception address
(IP A, port A) to a SIP proxy of a telecommunication network
(for example, of a cellular mobile radiocommunication network)
which applies SIP forking and, in steps 5 and 6, transmits

this SIP INVITE message to the called subscriber B terminal
(SIP terminal B) and the called subscriber B' terminal (SIP
terminal B').

Then, in step 7, the SIP terminal B selects its called
subscriber reception address (IP B, port B) and transmission
address (IP b, port b). In step 8 SIP terminal B' selects its
called subscriber reception address (IP B' and port B') for
reception, and its called subscriber transmission address (IP
b' and port b') for transmission.

In step 9 the called subscriber reception address (IP B, port
B) selected in called subscriber B and, according to the
invention, the called subscriber transmission address (IP b,
port b) together with a unique identification of the dialogue
B, are transmitted in a SIP 181 Ringing provisional response
message to a SIP proxy of a telecommunication network, which
transmits them to the calling subscriber (A) in step 10. In
addition, in step 11, a SIP 180 Session Progress provisional
response message with the further called subscriber reception
address (IP B', port B') and, according to the invention, the
called subscriber transmission address (IP b', port b') and
the dialogue identification B', are transmitted onwards by the
further SIP terminal B' to the SIP proxy and, (in step 12) to
SIP terminal A (the calling subscriber A).

To transmit the called subscriber transmission addresses (IP
b, port b) and (IP b', port b') in messages 9 to 12 according
to the invention, a newly-introduced SDP parameter may, for
example, be used.

Through the receipt of messages 9 and 11 with different
dialogue identifications B and B', the SIP terminal A
connection piece knows that it is signaling with two terminals
B and B', and that at this time both terminals are possibly

transmitting data (= early media data = media stream data) to
(IP A, port A), as in steps 13 and 14, from the called
subscriber (= SIP terminal B or B') to the terminal of the
calling subscriber A. At this happens the SIP terminal B (or
the further destination and SIP terminal B') specifies a
called subscriber transmission address (IP b, port b or IP b',
port b') indicating where the data originates, to enable the
calling subscriber A to determine its origin. In addition, the
early media data transmitted in step 13 and 14 also contains a
destination address of the calling subscriber (A) which is
used for IP routing. Early media data may contain, for
example, ring tones, announcements, etc.

If (through forking) calls are redirected to a plurality of
telecommunication network switching devices (proxies) and/or
SIP terminals (such as B, B') simultaneously or sequentially,
and are possibly redirected to further terminals by addressed
SIP terminals B, B' and/or proxies, provisional responses and
possibly early-media media stream data from many terminals may
arrive at terminal A of the calling subscriber, selection
among which is optimized simply and efficiently according to
the invention.

This is accomplished in that (contrary to the procedure
according to the standardization documents mentioned in the
introduction) a called subscriber B transmission address (IP
b, port b) is transmitted together with the called subscriber
reception address (IP B, port B) (transmitted in a response)
in a response message (provisional response or final response)
of a called subscriber B, and the called subscriber B
transmission address (IP b, port b) is used for the selection
(further processing or storage or rejection, etc.).

Rejection may take place, for example, if, after a "200-OK"
final response message has been forwarded by the called
subscriber terminal B to the calling subscriber terminal (A)
in steps 16, 17, and the successful ending of the call
initiation is signaled, so that an established dialogue
between terminal A and terminal B is then established,
whereupon, for example, early media data streams which do not
correspond to the established dialogue established with the
message 16/17 (and which therefore contain a different
subscriber transmission address) can be rejected (e.g.
suppressed or ignored) by the calling subscriber A. According
to the invention the suppression is effected in that the media
stream data with transmission addresses other than (IP b, port
b) is ignored. The SIP terminal A signaling part informs the
SIP terminal A connection part in message 17 that only media
stream data with the transmission address (IP b, port b) must
be accepted. For this purpose a new parameter expressing one
or more transmission addresses the packets of which must be
accepted is introduced, for example, in message 17. The same
new SDP parameter as is used in messages 9 to 12, which is
transported in SDP within a MOD message of the H.248 protocol,
may, for example, be used. Alternatively, the SDP parameter
proposed by the IETF MMUSIC Working Group in "draft-ietf-
mmusic-sdp-srcfilter" may be used.

In this way a so-called "clipping" can be avoided, i.e. a non-
existent user connection after initiation of the connection in
the signaling has been completed on the basis of a final
response of a SIP terminal B, once the user has gone off-hook.
The non-existent user connection is produced by further
processing of early media data streams which are no longer
relevant. Otherwise, it would be only upon receipt of a SIP
CANCEL message (step 20) from the SDP proxy to the further SIP

terminal (B'), for example, that (only) this SIP terminal B' would stop transmitting early media data streams, and the clipping could continue to exist in a transition period for as long as terminal A was still receiving this early media data.

For example, after receiving a message 9, 10 from B with called subscriber (B) transmission address data (IP b, port b) (contained in the message), the calling terminal A in the example in Fig. 1 can either select for further use user data (early media data) received later in a message 13 on the basis of called subscriber transmission address data (IP b, port b) contained in message 13, or reject, i.e. delete or ignore, user data containing called subscriber transmission address data which has not been forwarded in an earlier provisional response or final response message (9, 10; 11, 12).

The information contained in SDP on the source IP address and the source port is used by the terminal of caller A in the manner according to the invention to select packets from suitable media streams for representation.

In a preferred embodiment a new parameter which is used in the provisional responses and/or final responses sent by a terminal of B to the SIP terminal of A is introduced in SDP. This parameter enables the terminal(s) of caller B to express which IP address and which port is used in each case by these terminals to send IP packets. (Up to now, SDP from B to A contains only information on the IP address and port at which B wishes to receive IP packets.) Because the provisional responses and/or final responses contain a unique identification of the SIP dialogue, and the IP address used by a terminal B for transmission, and the port used for transmission, i.e. the source IP address and the source port,

in packets of the corresponding media stream received by A, it
is possible for A to carry out an unambiguous correlation
(allocation) between a SIP dialogue and a received media
stream.

The terminal of A uses this correlation to select suitable
media streams, e.g. as follows:
When the first early dialogue becomes an established dialogue
upon receipt of a SIP final response, the terminal of A
selects the appropriate media stream. And/or:
The terminal of A selects the early media corresponding to the
early dialogue last established - possibly for only as long as
no established dialogue yet exists. And/or:
The terminal of A suppresses early-media media streams (user
data) as soon as it sends SIP signaling messages to end the
corresponding dialogues.

The SIP terminal of A may be divided into a signaling device
and a device for handling user connections, which devices
communicate with one another, for example, by means of the
protocols H.248 or RFC 3525 jointly specified by the ITU-T and
IETF, or by means of the MGCP protocol IETF RFC 2705. For
example, the SIP terminal of A may consist of a 3GPP-
standardized MGCF and IM-MGW, or of a MRFC and MRFP, also
standardized by 3GPP, see 3GPP TS 23.002. SDP is also
transported by H.248 or MEGACO. Here, too, the new SDP
parameter according to the invention is used to specify the
source IP address and the source UPD port with which received
user connection packets are to be accepted. If this parameter
is used, user connection packets with other source IP
addresses and other source port are to be rejected.

Other examples of terminals which are not divided are mobile

3GPP or 3GPP2 terminals known as UEs.

For simplicity, various SIP messages, for example, 100 Trying,
PRACK and OK(PRACK) are not shown in Fig. 1.